LISTING OF THE CLAIMS

This listing of claims will replace all prior versions, and listings, of claims in the application:

1.-3. (Canceled).

4. (Previously Presented): A method for storing an electric signal representing recorded ambient noise in compressed form, the method comprising:

periodically recording samples of the ambient noise using a sound transducer,
the sample duration being shorter than the sampling cycle;
dividing the recorded audio signal into at least two band signals by filtering, with each one of the band signals containing a frequency range of the audio signal, and wherein any content of the other band signals contained in each band signal is present only in an attenuated form;
normalizing the amplitude of the divided audio signal within a first predetermined range D;
mapping the normalized amplitude values of the sampled ambient noise onto a second predetermined range of values in the time domain using a non-linear mapping function to obtain an emphasis of selected values ranges within the first or the second predetermined ranges; and
storing the mapped result in an electronic memory in a digital format.

5. (Previously Presented): The method of claim 4, wherein the audio signal is divided into from 3 to 15 band signals.

6. (Previously Presented): The method of claim 4, wherein the band signals essentially contain frequency ranges of the same width each, and all frequency ranges are comprised in the range of 500 Hz to 10,000 Hz.

7. (Previously Presented): The method of claim 4, wherein the band signals are generated by splitting once or a cascaded multiple of times an input signal which is either the audio signal or an output signal obtained according to the following steps:

first low pass filtering to generate a first output band signal, and
subtracting the first output band signal from the input signal to generate a second output band signal.

8. (Previously Presented): The method of claim 7, wherein the low pass filtering is realized by means of a digital convolution over 10-30 values.

9. (Canceled)

10. (Previously Presented): The method of claim 7, wherein the input signal is digitized and only every nth value of each division stage is added to the band signal, n being greater than or equal to 2, in order to compensate for the increased data volume resulting from the splitting into band signals.

11. (Previously Presented): A method for storing an electric signal representing recorded ambient noise in compressed form, the method comprising:

periodically recording samples of the ambient noise using a sound transducer, the sample duration being shorter than the sampling cycle;

normalizing the amplitude of a signal output of the transducer or a signal derived therefrom within a first predetermined range D;

mapping the normalized amplitude values of the sampled ambient noise onto a second predetermined range of values in the time domain using a non-linear mapping function to obtain an emphasis of selected values ranges within the first or the second predetermined ranges; and

storing the mapped result in an electronic memory in a digital format; and

generating an energy signal which is proportional to an energy content of the ambient noise from the audio signal or from a signal derived from the audio signal.

12. (Previously Presented): The method of claim 11, wherein the energy signal is subjected to a second low pass filtering.

13. (Previously Presented): The method of claim 12, wherein the second low pass filtering is effected digitally in the form of a convolution over 20 to 70 values.

14. (Previously Presented): The method of claim 13, wherein the second low pass filtering is followed by a second data reduction where one energy value among n filtered values is selected, n being at least equal to 2.

15. (Previously Presented): The method of claim 11, further comprising performing a subsequent differentiation of the energy signal with respect to time to obtain an energy difference signal.

16. (Previously Presented): A method for storing an electric signal representing recorded ambient noise in compressed form, the method comprising:

periodically recording samples of the ambient noise using a sound transducer,

the sample duration being shorter than the sampling cycle;

normalizing the amplitude of a signal output of the transducer or a signal derived therefrom within a first predetermined range D;

mapping the normalized amplitude values of the sampled ambient noise onto a second predetermined range of values in the time domain using a non-linear mapping function to obtain an emphasis of selected values ranges within the first or the second predetermined ranges; and

storing the mapped result in an electronic memory in a digital format; wherein:

the range of normalized values D is defined by a lower limit D_u, and an upper limit D_o, and the normalization is effected by:

obtaining the maximum of the absolute value of the audio signal or the derived signal within the duration of normalizing the audio or derived signal, which is shorter than or equal to the duration of a hearing sample,

multiplying the reciprocal value of said maximum by (D_o - D_u +1), and

multiplying this product by each value of the audio or derived signal within the duration of the normalized signal.

17.-32. (Canceled)

33. (Previously Presented): The method of claim 4, wherein any content of the other band signals contained in each band signal is attenuated to half of their respective original levels.

34. (Previously Presented): The method of claim 4, wherein any content of the other band signals is completely attenuated from each band signal so as to not be present at all therein.

35. (Previously Presented): The method of claim 5, wherein the audio signal is divided into from 4 to 10 band signals.

36. (Previously Presented): The method of claim 5, wherein the audio signal is divided into from 5 to 8 band signals.

37. (Previously Presented): The method of claim 5, wherein the audio signal is divided into 6 band signals.

38. (Previously Presented): The method of claim 7, wherein all first low pass filterings have a same Q-factor.

39. (Previously Presented): The method of claim 8, wherein the low pass filtering is realized by means of a digital convolution over 15 to 25 values.

40. (Previously Presented): The method of claim 8, wherein the low pass filtering is realized by means of a digital convolution over 19 values.

41. (Previously Presented): The method of claim 10, wherein n is equal to 2.

42. (Previously Presented): The method of claim 11, wherein the energy signal is generated by squaring said audio signal or said signal derived therefrom.

43. (Previously Presented): The method of claim 13, wherein the second low pass filtering is effected digitally in the form of a convolution over 40-55 values.

44. (Previously Presented): The method of claim 13, wherein the second low pass filtering is effected digitally in the form of a convolution over approximately 48 values.

45. (Previously Presented): The method of claim 13, wherein the convolution has coefficients which are essentially equal to each other.

46. (Previously Presented): The method of claim 13, wherein the coefficients of the convolution are equal to 1.0.

47. (Previously Presented): The method of claim 14, wherein n is equal to the number of values of the convolutions of the second low pass filtering.

48. (Previously Presented): The method of claim 15, wherein the differentiation is performed by computing the difference between two respective values of the energy signal.

49. (Currently Amended): The method of claim 16, wherein [[Do]] $\underline{D_u}$ is equal to 0.

50. (Currently Amended): The method of claim 16, wherein $\cancel{D_u - D_0}$ $\underline{D_0\text{-}D_u}$ is equal to $\cancel{2_n - 1}$ $\underline{2^n\text{-}1}$, n being a whole number greater than 4.

51. (Previously Presented): The method of claim 50, wherein n is equal to 7.

52. (Previously Presented): The method of claim 16, wherein the duration of normalizing the audio or derived signal is equal to the duration of a hearing sample.

53.-60. (Canceled)

61. (Previously Presented): The method of claim 8, wherein for the purpose of the low pass filtering, the convolution is performed according to the relationship:

$$y_j = \sum_{i=0}^{18} a_i * x_{j-i}$$

where:

j is the time index, y_j is the output value of the low pass filtering at the time j;

x_j is the input value for low pass filtering at the time j;

a_i is the coefficient of the convolution sequence; and

a_0-a_{18} are [0.03, 0.0, -0.05, 0.0, 0.06, 0.0, -0.11, 0.0, 0.32, 0.50, 0.32, 0.0, -0.11, 0.0, 0.06, 0.0, -0.05, 0.0, 0.03]